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ACTIVE NOISE CONTROL SYSTEM IN UNRESTRICTED SPACE

FIELD OF INVENTION

The present invention relates to a noise control system, which is preferably an active noise control system, and a method for controlling noise, particularly but not exclusively in large unrestricted spaces.

BACKGROUND OF INVENTION

Conventional adaptive cancellation systems using traditional transverse finite impulse response (FIR) filters, together with least mean square (LMS) adaptive algorithms, well known in the prior art, are slow to adapt to primary source changes. This makes them inappropriate for cancelling rapidly changing noise, including unpredictable noise such as speech and music. Secondly, the cancelling structures require considerable computational processing effort to adapt to primary source and plant changes, particularly for multi-channel systems.

A general structure for such a cancellation system is shown in the applicant's international application having publication no. WO 01/63594. Here a primary source to be cancelled, a cancelling secondary source and an error sensor are in successive substantial alignment. Noise emanating from the primary source is cancelled using the second noise source and optimum cancellation is achieved by measuring the error between the unwanted primary noise and the actual noise produced by the second source. This error is fed to a system of FIR filters as a feedback for adjusting the noise produced by the second noise source.

These FIR filters adapt with increasing speed (reduced time constant) in reducing the noise, as the number of transverse control taps (coefficients) in the filter is increased to an optimum value. The adaptive speed at which the cancellation noise adapts to match the unwanted noise increases with the cancelling strength $\beta=\mu A^2$, where μ is the adaptive step size of the cancellation noise with each adaptive iteration and A is the peak signal amplitude of the cancellation noise. The speed also decreases with increases in the spectrum density. Thus for a primary source with frequencies of various amplitudes, the adaptive speed will reduce as the number of source frequencies increases, with the lower amplitudes adapting more slowly. If the signal is non-varying, then the lower amplitude frequencies will adapt

eventually, given sufficient taps and time. But for source frequencies varying in time the smaller amplitudes will not have time to catch up (adapt completely), producing slow adaptation and signal distortion.

Further disadvantages of the conventional transverse FIR adaptive systems are (i) basic instability, where the error sensor is permanently required and functioning to maintain stability (ii) deteriorated cancellation away from the error sensor and (iii) susceptibility to environmental changes, through a large controlling propagation distance.

SUMMARY OF THE INVENTION

According to a first aspect of the present invention there is provided a noise control system as set out in claim 1.

According to a second aspect of the present invention there is provided a noise control system as set out in claim 7.

Various preferred or optional features are defined in the other claims.

BRIEF DESCRIPTION OF DRAWINGS

Embodiments of the invention will now be described, by way of example only, with reference to the accompanying drawings in which :

Figure 1 is a block diagram of a multi-bandpass, variable μ, fixed β, transverse FIR adaptive filter, in accordance with a first embodiment;
Figure 2 is a block diagram of an instantaneous plant inverse negative direct replica cancelling (IPINDR) system, in accordance with a second embodiment;
FIG 3 is a diagram illustrating signal alignment in sample numbers; and
FIG 4 is a block diagram of multi-channel configurations using the IPINDR approach of the second embodiment.

DESCRIPTION OF PREFERRED EMBODIMENTS OF THE INVENTION

In the figures, like reference numerals indicate like parts and multiple like elements are denoted using lower case letters as sub-reference numerals.

Referring firstly to figure 1, there is shown a multi-passband, variable μ, fixed β method to increase the adaptive speed of transverse FIR filters to primary source changes. The approach is to divide the source spectrum bandwidth into frequency pass-bands, where each passband has a separate FIR filter with its own μ made inversely proportional to A^2 in each passband, tending to maintain a constant β and therefore adaptive speed, irrespective of the spectrum amplitude. Thus a faster and similar response of the cancellation sound to the unwanted noise, will be more nearly obtained as the number of passbands increase for a given total spectrum bandwidth.

Figure 1 shows a primary noise source 1 which produces a primary noise to be cancelled. This noise is shown to propagate along a primary path 2. There is further shown a primary transducer in the form of a microphone 4 disposed close to the primary noise source 1, and arranged to feed measured reference sound 'x' into a control box 12. An output from the control box 12 is arranged to be fed to a speaker 7, which produces a secondary cancelling sound that passes along a secondary path 8. Noise from both the primary and secondary paths is arranged to be received by an error transducer in the form of a microphone 3. The output error signal E of this microphone 3 is fed into the control box 12.

In operation, the noise from the primary source 1 propagates along the primary path 2 to be received at the error microphone 3. The noise is also measured in close proximity to the primary source 1, using the microphone 4. The resulting secondary signal x from the microphone 4 representative of the primary source noise is then fed into the control box 12.

Within the control box 12 there are provided a number n of pass band and filter arrangements. Only the first two of these is shown in detail, and a variable number of further arrangements can be added as required, as will be explained below. The first of these two arrangements comprises a passband 1, labelled with reference numeral 5a, a conventional finite impulse response (FIR) 1 filter 6a, a conventional control system transfer function estimate 9a including estimates of the elements 4,7,8 and the computational implementation (not shown), a conventional least mean square (LMS) or its equivalent algorithm 1 10a and an adaptive step size 11a.

The second passband and filter arrangement comprises corresponding elements labelled with the sub-reference numeral b. Similarly, each of the n arrangements has corresponding "n" elements.

The above-mentioned secondary signal from the microphone 4 is passed into each of the n passband filters 5n. The process will be described with reference to the first passband and filter arrangement. Thus the secondary signal is passed into passband 1 5a, and an output from this filter is passed through the FIR 1 filter 6a, to the secondary transducer, loud speaker 7. The loud speaker 7 generates the secondary cancelling sound that propagates through the secondary propagation space 8 to the error microphone 3, as mentioned previously.

The output from the passband 1 filter 5a is also passed through the control system estimate 9a and the output of the plant estimate 9a is then passed into the least mean squared LMS 1 algorithm 10a. Also fed into the LMS 1 algorithm 10a is the error signal E from the error microphone 3 and the adaptive step size 11a, which is automatically calculated from the passband 1 5a output level such that the adaptive step size is adjusted proportional to A^2 with each adaptive time step. The output from the LMS algorithm 10a is passed into the FIR filter 1 to control the FIR 1 filter 6a adaptive process so as to drive that part of the error signal E caused by the pass-band 1 to a minimum.

Similarly, the output from the primary microphone 4 is passed into the passband 2 filter 5b, through the FIR 2 filter 6b into the secondary loud speaker 7. The loud speaker 7 generates the secondary sound that propagates through the secondary propagation space 8 to the error microphone 3. The output from the pass-band 2 filter 5b is passed through the same control system estimate 9b, then into the LMS 2 algorithm 10b, together with the error signal E from the error microphone 3 and the output from the automatic adaptive step size 11b, whose size is determined by the output from passband 2 filter 5b. The output from the LMS algorithm 10b then controls the FIR 2 filter 6b adaptive process to drive the error signal in its passband to a minimum.

To extend the total frequency bandwidth or reduce the spectrum energy per passband, additional 'n' passband adaptive systems, each equalizing the adaptive speed in each of its passbands, can be added. The number of passbands will therefore depend on the spectrum

density, the total spectrum bandwidth and the speed of adaptation to variations in the noise x required.

As the adaptive strength β and therefore speed, is proportional to the peak signal amplitude 'A' squared times the adaptive step size μ in each passband, then if the step size is reduced proportional to the signal amplitude squared, automatically, then the adaptive strength β will be maintained within the passband irrespective of amplitude.

Applying the same technique in each passband will tend to give an equal response to all frequencies in all the passbands. This increases the overall adaptive speed and reduces the spectrum distortion compared with a conventional transverse FIR filter. In other words, this embodiment increases the adaptive speed of the system to cancel the primary noise evenly across the frequency spectrum as the primary noise varies, thus reducing the signal distortion. However, the method has a maximum adaptive speed limited by a finite cancelling strength β. As β increases the stability bandwidth shrinks, its maximum value is given by the stability zero band width, as considered by Wright et al, Journal of Sound and Vibration (2001) 245(4).

The approach of the embodiment of figure 1 is therefore an improvement over prior art systems and is adequate for moderately changing primary sources such as unsteady periodic noise. It can have the disadvantage of intensive computation as it requires adaptive FIR filters and FIR passband filters for each band, although the passband filters could be implemented into hardware to reduce the computational burden.

To implement a really fast response to source changes, including unpredictable noise, and avoiding the disadvantages of the first embodiment, the online adaptive transverse FIR filters are removed and the primary source signal cancelled with a negative copy of itself, directly.

A time domain solution that gives virtually instantaneous response to primary source changes and is computationally efficient, is to negate a copy of the primary source signal, compensate for signal distortion caused through hardware implementation of the secondary cancelling system, align and match the resulting secondary wave with the primary wave at its instantaneity point.

A second embodiment of the invention, as shown in figure 2 is arranged to achieve these advantages and mitigate the disadvantages of the first embodiment of figure 1.

To increase the response to rapidly changing primary sources, to avoid the disadvantages of conventional adaptive FIR filters discussed earlier, and to reduce the computational effort, the control box 12 in Figure 1 is replaced with a control box 18 in Figure 2. The process used to deal with the cancellation of arbitrary noise, including non-periodic unpredictable noise, is described generally in the time domain. Again to generate the secondary cancelling signal, a copy x of the primary source signal is measured using the primary microphone 4.

The control box 18 contains a negator 13, a control system neutralisation inverse estimate 14, an inverse delay required to obtain the inverse system estimate 15, an amplitude control 16 and an adjustable sample delay buffer 17, all arranged in series. The error signal E from the error microphone 3 is passed into each of the attenuation regulator 16 and the adjustable sample delay 17.

In operation, the output from the primary microphone 4 is negated in negator 13, and then convolved with the control system neutralization inverse estimate 14, which removes the signal distortion produced by the cancelling system hardware. The control system inverse 14, for example, in the form of an FIR filter can be measured directly in series with the control system. For non-minimum phase inverse functions the delay n_{inv} 15 is used in parallel with the control system and its inverse to realize these functions. This delay effectively becomes part of (series with) the inverse system estimate.

An alternative is to determine the system inverse from its impulse response measured in parallel with the control system. Then the inverse can be obtained through the frequency domain, as described below under the heading "Inverse Functions".

The signal is then passed through the amplitude control 16 and the adjustable delay buffer 17, and then to the secondary loud speaker 7, where the resulting secondary signal Y propagates through the secondary propagation space 8, arriving at the error microphone 3 as Y_s'. The signal from the primary source passes along the primary path 2 to the error microphone 3 as before, and is labelled in figure 2 as Y_p'.

Details of operation and characteristics of the system of figure 2 will now be described.

Cancelling Characteristics

The instantaneous, plant inverse, negative direct replica (IPINDR) system has the following characteristics:

1) Secondary cancelling signal is 'copied' from the primary source using a primary sensing transducer (microphone or equivalent), suitably isolated from the secondary source (shielding and/or directional transducers) to prevent feedback between the two.

2) Secondary signal is negated in preparation for cancelling the primary signal.

3) The electromechanical system (impulse) response I_{em}, which produces distortion in the cancelling signal, is neutralised/reduced by (i) physically altering the dynamic response of the system, particularly the dominant component, namely the sound transducer (the loud speaker 7) together with its power amplifier (not shown), (ii) mathematically modifying the net response of the system through adding the appropriate poles /zeros to the overall transfer function, (iii) measuring the impulse response of the system and inverting. The system includes essential components in the secondary sound cancelling path (computer A/D, D/A converters, aliasing/quantisation filters, amplifiers microphones and loudspeakers)

4) The physically modified control system, and/or the neutralised control system is used to drive the secondary source (i.e. the cancelling loud speaker 7).

5) The resulting secondary acoustic wave is combined and aligned with the primary acoustic wave by appropriately positioning the secondary source 7 downstream of the primary source 1 in the direction of the wave propagation and the error microphone 3. This facilitates a time advance relative to and along the primary wave represented by the shift function $h(t+\tau_a)$, where τ_a is the time advance and is given by $\tau_a = r_{ps}/c_o$.

6) The time advance τ_a is the wave propagation time between the primary 1 and secondary 7 sources, r_{ps} is the propagation distance between the sources and c_o is the propagation speed (speed of sound). The time advance is necessary to offset the cancelling signal processing delay represented through $h(t-\tau_r)$, where τ_r is the secondary path processing time retardation.

7) The cancellation is dependent on the distance between the primary 1 and secondary 7 sources, $r_{ps}= r_{pm}-r_{sm}$ (r_{pm} is the primary source microphone-error microphone distance), not on the secondary source-error microphone distance r_{sm}, as in the case of the conventional prior art adaptive FIR cancelling process. The controlling distance r_{ps} is considerably smaller than the controlling distance r_{sm}. This makes this critical propagation space much less vulnerable to environmental changes, such as fleeting reflections, than in the conventional adaptive FIR method.

8) Acoustically the primary and secondary sources form a phase controlled dipole (PCD), as described in Journal of Sound and Vibration (2001) 245 (4). Here the phase of the secondary source is adjusted to be out of phase with the primary sound field at the error microphone 3 located downstream in successive alignment following the primary 1 and secondary 7 sources. The resulting radiated acoustic field directivity (shadow shape) can be adjusted to be progressively tripole (cardiod), dipole (figure of eight) and quadrupole (four leaf clover), as the difference between the primary 1 and secondary 7 source distance r_{ps} increases.

9) The PCD, in this direct negative replica case, uses the propagation distance r_{ps} for both the primary and secondary waves. This produces exact alignment between the waves, giving maximum shadow at all points along the wave from the primary source, in the direction of the error microphone 3. By contrast, in the conventional adaptive FIR system, the propagation distance r_{pm} is used for the primary path and r_{sm} for the secondary path. This produces exact alignment only at the error microphone, giving a slight phase difference at all other points along the wave, progressively deteriorating the shadow with distance.

10) The IPINDR cancelling system of figure 2 is inherently stable requiring the error microphone 3 only to set up the cancellation process. After the setting up, the cancellation is self-sustaining, without the use of the microphone 3, except for all but severe environmental changes.

Mathematical Description

Equivalent control concepts apply equally to analogue systems, but modern digital systems are more precise and do not suffer from drift. The control is therefore described in terms of digital control. For these systems, the control is implemented through samples generated by

the sampling frequency f_n. The time advance $\tau_a = r_{ps} / c_o$, where c_o is the speed of sound, is equivalent to a sample advance number of:

$$n_a = \tau_a f_n = r_{ps} f_n / c_o \qquad (1)$$

The total sample delay (retardation) n_r is generated through (i) the unavoidable secondary control system implementation time delay n_{imp}, including the control system inverse delay n_{inv} needed to retard advanced inverse functions (as calculated in the control system delay 15) and (ii) an adjustable sample delay n_b intentionally added through the delay buffer 17 (or equivalent means) to fine tune off line, or momentarily on line, signal alignment, particularly through considerable environmental changes.

This gives a total sample retardation number:

$$n_r = n_{imp} + n_b, \qquad n_{imp} \approx n_{inv} \qquad (2)$$

For a periodic wave, the secondary wave alignment with the primary wave (as illustrated in Figure 3 to be described in more detail below), is given by:

$$N_p n_p - \Delta n = 0, \qquad \Delta n = n_r - n_a, \qquad n_p = T_p / T_n = f_n / f_p \qquad (3)$$

Where n_p is the number of samples in the period T_p of the primary wave of periodic frequency f_p and N_p is the period number that the primary wave is in advance of the secondary wave giving:

$$n_a = n_r - N_p n_p \qquad (4)$$

For a slowly changing periodic noise the system can be non-causal i.e. the delay τ_r can be longer than the advance τ_a, as here only the periods need to be aligned i.e. N_p can be any integer. For unpredictable noise the signals must be causal and exactly aligned, and the advance must balance the delay exactly, i.e. N_p=0, making

$$h(t+\tau_a)h(t-\tau_r) = h(t + \tau_a - \tau_r) = h(t) \qquad (5)$$

The sample advance n_a is adjusted by adjusting the distance between the primary and secondary source r_{ps}, according to equation (1), until n_a is approximately the same as but greater than n_r. The delay buffer n_b in equation (2) is then fine tuned until $n_a = n_r$, giving a minimum error E at the error microphone 3. The amplitude A of the secondary signal is adjusted to match that of the primary source signal giving a minimum error E at the error microphone 3.

The last two steps are successively repeated, manually or automatically, until the lowest minimum error E is achieved. This indicates that the secondary and primary signals are in alignment at the error microphone 3, and at all points along the wave.

Correlation Process

Referring back to figure 2, the difference between the primary signal $Y_p'(t)$ and secondary signal $Y_s'(t)$ at the error microphone becomes:

$$E'(t) = Y_p'(t) - Y_s'(t) = X(t) \star [P_{ps} \star P_{sm} - I_{em} \star (I_{em}{}^*)^{-1} \star S_{ps} \star S_{sm}] \qquad (6)$$

Where $\star$ indicates linear convolution, x(t) is the reference signal at the primary source, P_{ps} and P_{sm} are the primary path responses, i.e. primary to secondary source and secondary source to microphone, respectively. I_{em} is the actual electro-mechanical control system impulse response of the cancelling system and $(I_{em}{}^*)^{-1}$ is the measured or calculated inverse of the electromechanical control system impulse response. S_{ps} and S_{sm} are the primary–secondary source computation delay and secondary source-microphone path responses, respectively.

If the propagation path terms $P_{sm} = S_{sm}$, and $P_{ps} = h(t-\tau_a)$ is a pure delay, and further the total computation delay $S_{ps} = A.h(t-\tau_r)$ where A is an amplitude adjustment, then the difference signal at the secondary loud speaker becomes:

$$E(t) = Y_p(t)-Y_s(t) = X(t) \star S_{sm}(t) \star [h(t-\tau_a) - A.\ I_{em} \star (I_{em}{}^*)^{-1} \star h(t-\tau_r)] \qquad (7)$$

For a time varying periodic noise or unpredictable noise, the signals have to be matched exactly. Thus the zero order period $N_p=0$ has to be used giving $n_a = n_r$ and $\tau_a = \tau_r$.

In this case, equation (7) becomes in the frequency domain:

$$E(f) = Y_p(f) - Y_s(f) = X(f)\, S_{sm}(f)\, h(f)\, [1 - A.B(f)/B^*(f)\, e^{j(\theta-\theta^*)}] \qquad (8)$$

Where f is the acoustic frequency, $h(f) = e^{j2\pi f \tau r}$, B and B* are the amplitudes and θ and θ* are the phases of the impulse response I_{em} and estimated (measured) response I_{em}^* respectively. For zero frequency distortion, the plant dynamics has to be neutralised completely, from equation (8)

$$n_a = n_r,\quad A = B^*/B,\quad \theta^* = \theta \text{ giving } E=0 \qquad (9).$$

There is a minimum distance r_{ps} between the primary and secondary source for cancellation to be achieved. This is determined by the secondary path processing time which is basically the delay n_{inv} required in the inverse function realization. From equations (1), (2) and (9) this distance is given by

$$r_{ps} = n_{inv}\, c_o / f_n \qquad (10)$$

This is the minimum distance for the cancellation of unpredictable noise to succeed. n_{inv} can be large for non minimum phase control system functions.

Thus the secondary signal Y_s' is aligned with the primary signal Y_p', initially by adjusting, approximately, the distance r_{ps} in equation (10), and then fine tuning by adjusting the sample delay buffer n_b 17 to give minimum error E at the error microphone 3. The amplitude of the secondary signal is matched to that of the primary signal by adjusting the amplitude at the amplitude adjustment 16, to give a minimum error at the error microphone 3. The amplitude A and the delay n_b are then successively adjusted until a minimum error is achieved at the error microphone 3, manually or automatically.

Referring again to figure 3, this figure illustrates the secondary signal alignment with the primary signal in sample numbers. The primary source 1 is shown to produce a primary wave 21 of period T_p propagating rightwards in the figure, where n_p is the number of samples in the period T_p and N_p is the period number that the primary wave 21 is in advance of a secondary wave produced from the secondary source 7. The secondary wave position as measured from the primary microphone 4 and outputted directly from the loudspeaker 7,

without any delay between the primary microphone 4 and the loudspeaker 7 is shown by the dashed representation 22.

Moving the speaker rightwards in the figure by n_a samples also moves the secondary wave with it and advances its time compared to the primary wave 21.The position of the secondary wave after including a processing delay n_r is shown by the solid representation 23. For cancelling steady periodic noise the periods need only to be aligned (N_p integer in equation (4)). For unpredictable noise the secondary signal needs to be aligned exactly with the primary signal (N_p=0). This is accomplished by adjusting the propagation distance between the secondary loudspeaker 7 and the primary source n_a to equal that of the computation delay n_r, making Δn=0 in equation (3).

Shadow Bending

Shadows are formed at an angle α_B from the line joining the primary 1 and secondary 7 sources, from equation (1)

$$n_B = \Delta r_{ps}\, f_n/c_o, \qquad \Delta r_{ps} = r_{ps} - r_{ps}' = r_{ps}(1 - \cos\alpha_B) \qquad (11)$$

where n_B is the buffer sample change, r_{ps}' is the propagation distance in the direction of the shadow minimum. Rearranging the above equation gives

$$\alpha_B = \cos\text{-}1[(f_n r_{ps} - n_B c_o) / f_n r_{ps}] \qquad (12)$$

The shadow bending or rotation from the source axis, per n_B, therefore depends on the relative magnitude $f_n r_{ps}$ compared to c_o.

Inverse Functions

To obtain minimum distortion of the cancelling process, resulting in maximum cancellation, it is important to implement neutralisation of the secondary control system response. This can be obtained through an accurate measurement of the inverse of the actual electromechanical control system impulse response $(I_{em})^{-1}$.

An estimate of $(I_{em}*)^{-1}$ can be obtained in the time domain, directly in series with the actual I_{em}, off-line, using a white noise training signal. Care is needed in performing direct inverse estimates, as inverted functions are potentially unstable. For example, proper functions (functions with more poles than zeros) become improper functions when inverted. More seriously, 'unstable' zeros lying outside the unit circle in the Z domain (non-minimum phase functions) become unstable poles, turning delays into advances, when inverted. For these advanced functions in negative time to be to realized (i.e. for the adaptive process to converge effectively), a delay n_{inv} is required in parallel with the training process to delay these functions into real (positive) time.

A method that does not require a training delay is to obtain the inverse directly from the impulse response. An estimate $I_{em}*$ is measured in parallel with the actual I_{em}, using a white noise training signal. The spectrum amplitude B and phase θ are then obtained through performing the discrete fast Fourier transform (FFT) or swept spectrum or equivalent on $I_{em}*$ thus:

$$\mathrm{FFT}\,(\, I_{em}*) = \Sigma\,(Be^{j\omega\theta}) \qquad (13)$$

The inverse is then obtained by simply inverting B and negating θ, and then reassembling them back into the time domain, thus the inverse fast Fourier transform (IFFT) becomes :

$$\mathrm{IFFT}\,\Sigma\,(Be^{j\omega\theta}) \;=\Sigma\,(B^{-1}e^{-j\omega\theta)} = (I_{em}*)^{-1} \qquad (14)$$

A delay to retard the function can be added later as required.

<u>Multi-Channel Systems</u>

A single channel PCD cancelling system produces a narrow cancellation region (shadow). For practical systems requiring wide shadows, particularly at high frequencies, multi-channel (multi-secondary source - multi error detector) systems are required, to generate a practical shadow over a wide well defined angle. The primary source microphones, secondary cancelling sources and error microphones are generally arranged in successive planes or arcs from the primary source and contained within defining control angles, forming boundaries for the acoustic shadows, as described in International publication no. WO 01/63594.

For these multi-channel systems to operate effectively, the sound propagation path differences (Δr_{pd}) between the various combinations of cancelling speakers and error microphones of multiples (p) of acoustic half wavelengths ($\lambda_p=c_o/f_p$ where f_p is a series of frequency peaks) should be avoided, which is also described in International Publication no. WO 01/63594, giving

$$\Delta r_{pd} = pc_o/2f_p \quad \text{where} \quad p = 1,2,3,\ldots\text{etc.} \qquad (15)$$

Or in terms of sample numbers, for a discrete sampling system, $n_{pd} = t_{pd}\, f_n$ where $t_{pd} = \Delta r_{pd}/c_o$, the uncontrollable sample numbers and uncontrollable frequencies to be avoided are

$$n_{pd}=pf_n/2f_p \qquad \text{or} \qquad f_p=pf_n/2n_{pd} \qquad (16)$$

Generally, IPINDR multi-channel systems are fundamentally stable i.e. they do not require the error microphone to maintain cancelling stability. The cancelling system is basically instantaneous to the response of primary source changes, as a negative copy of the primary source signal is passed directly through the secondary source system to the cancelling loud speaker. Apart from the convolution, there are no computational demanding processes either. A simple phase and amplitude error adjustment is effected using a simple delay buffer and amplitude regulator.

Therefore, for a non-changing control system, the error microphone can be dispensed with after the initial setting up to produce minimum error (sound). Each channel can be set up independently, requiring no inter-channel coordination. Of course a multi-channel computer coordinated system should always out-perform a set of independent channels.

Figure 4 shows four possible configurations. Although these configurations are shown with respect to the second embodiment (IPINDR system), they could be used with respect to the first embodiment (of figure 1) with the exception that each channel requires a permanent error microphone. In this case, where the control boxes 18 and 21 are shown, control box 12 would be substituted.

Figure 4(a) shows the configuration for a small or large in-phase primary source 1 generating a shadow over an angle 19. Here a single primary microphone 4 is sufficient to drive all the

secondary sources 7. A single error microphone 3 is sufficient to adjust each channel, one at a time, at each of the angle positions, as indicated with the dotted outline. Within the adjustable control boxes 18 are the adjustment control elements including the amplitude regulator A and the delay buffer n_b shown in the chain dotted box 18 in figure 2. The secondary sources 7 and error detectors 3 are arranged generally in successive planes or arcs from the primary source and contained within control angles 19 forming shadow angles, both horizontally and vertically (not shown).

Figure 4(b) is a configuration for an out of phase primary source 1 (for example modal distributions within a metal structure). Here separate primary microphones 4 are used to measure the local sound variations across the primary source and drive each channel separately, making them self-contained units. Each unit consists of a primary microphone 4, control system 18, and loud speaker 7. Again only a single error microphone is used in turn, at each angular position, to minimise the error signal for each channel, one at a time and then as a group.

The amplitude A and delay n_b adjustments in control box 18 can be coordinated through computer control to align channels to give a collective minimum error at the error sensors for off-line adjustment, or momentary on-line adjustment for severe environmental changes. These control elements can also be replaced with, for example, a simple C filter (few taps FIR transverse filter and a modified filtered x algorithm), as in the control box 21 (see below).

Figure 4(c) shows such a computer coordinated multi-channel system. An array of units 4, 18 and 7 and an array of permanent error microphones 3 are shown in full line. Each of the error microphones 3 and control boxes 18 is linked to a computer 20. The control elements, amplitude A and delay n_b, in control box 18, are adjusted automatically through the computer 20 to produce a minimum collective error at the error microphones 3.

Sound propagation path differences between the secondary sources 7a,7b,7n and the error detectors 3a,3b,3n of multiples of acoustic half wavelengths should be avoided for these multi-channel systems to operate effectively, as described in equations (15) and (16) above. All configurations are capable of shadow angle rotation through appropriate adjustment of n_B, r_{ps} or f_n.

Figure 4(d) shows the details of a further example of a computer-adjusted system. The control box 18 is replaced with the control box 21. Element 22 is the measured control system inverse, element 23 is the inverse delay required to obtain the inverse, element 24 is a fine adjustment C filter (low order FIR transverse filter) and element 25 is the impulse response of the secondary path r_{sm} and control elements 22, 23 and 7.

The impulse response filters the reference signal x, from the primary microphone 4, before it is used in the adaptive algorithm 26 to align the primary and secondary waves. The adaptive algorithm 26 also uses the output from the error microphone 3. n_{ps}, n_{sm}, and n_{pm} are propagation distances in sample numbers between the primary source - secondary source 7, the secondary source 7 - error microphone 3, and the primary source 1 - error microphone 3, respectively. The relationships between propagating distances in samples and the secondary control system impulse response l_{sm} , where z is the z domain discrete time transform, are :

$$n_{ps}+n_{sm} = n_{pm} \quad \text{and} \quad l_{sm} = l_{em}\, z^{-nsm}, \qquad (17)$$

giving the filtered x impulse response l_x as

$$l_x = (l_{em}{}^*)^{-1}\, z^{-ninv}\, l_{sm} = (l_{em}{}^*)^{-1}\, l_{em}\, z^{-(nsm+ ninv)} \qquad (18)$$

If $l_{em}{}^* = l_{em}$

$$l_x = z^{-(nx)}, \quad n_x = n_{sm}+ n_{inv}, \quad \text{and} \quad n_c = n_{pm}- n_x= n_{pm}- n_{sm} - n_{inv} = n_{ps} - n_{inv} \qquad (19)$$

This adjustment scheme is practically instantaneous. If the control system inverse estimate is accurate i.e. $l_{em}{}^* = l_{em}$, then the filtered x delay n_x becomes simply the sum of the secondary path delay n_{sm} and the inverse delay n_{inv} and the C filter delay n_c becomes the difference between the primary secondary source delay n_{ps} and the inverse delay n_{inv}. For unpredictable noise $n_c>0$ making $n_{ps}> n_{inv}$. For predictable noise the C filter can also be used to reduce the minimum distance r_{ps} in equation (10). As the number of taps W_c decreases, the adaptive step size μ_c increases.

The applicant draws attention to the fact that the present invention may include any feature or combination of features disclosed herein either implicitly or explicitly or any generalisation thereof, without limitation to the scope of any of the present claims.